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(54) **INTEGRATING MULTIMEDIA CAPABILITIES WITH CIRCUIT-SWITCHED CALLS**

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(52) **U.S. Cl.** **370/352; 370/315; 370/356; 370/389; 370/401; 370/351; 379/201.07**

(58) **Field of Classification Search** **370/401, 370/389, 410, 351-356, 315; 379/201.07**
See application file for complete search history.

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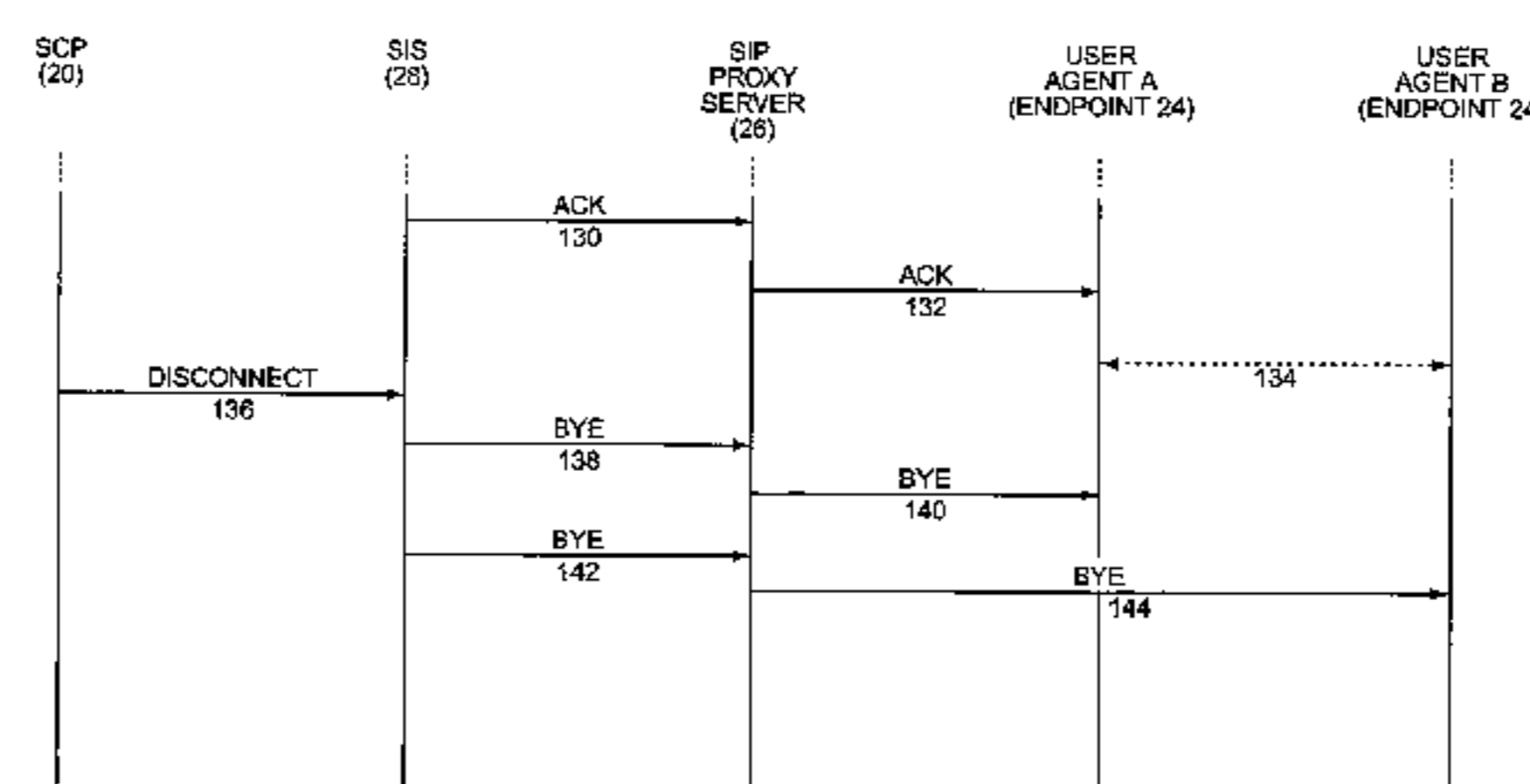
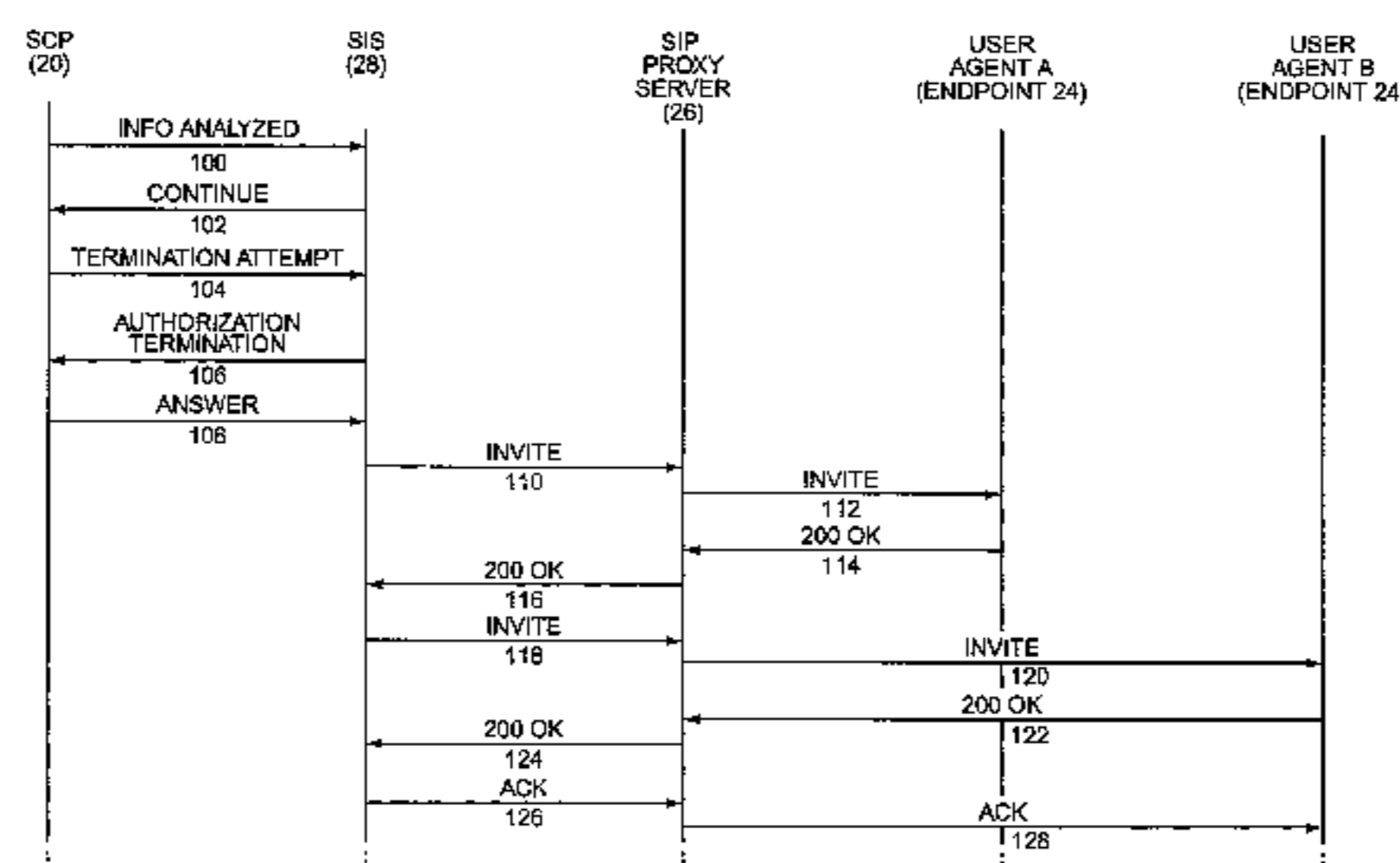
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(57) **ABSTRACT**

The present invention monitors call signaling events stemming from a circuit-switched call between a caller and a called party and controls a packet-session between user agents on respective endpoints associated with the caller and called party. The endpoints may include any type of computational device capable of facilitating the packet-session over a packet-switched network. Control of the user agents may be provided via a proxy for the user agents and may use the session initiation protocol (SIP), or like session control protocol for communications.

27 Claims, 4 Drawing Sheets



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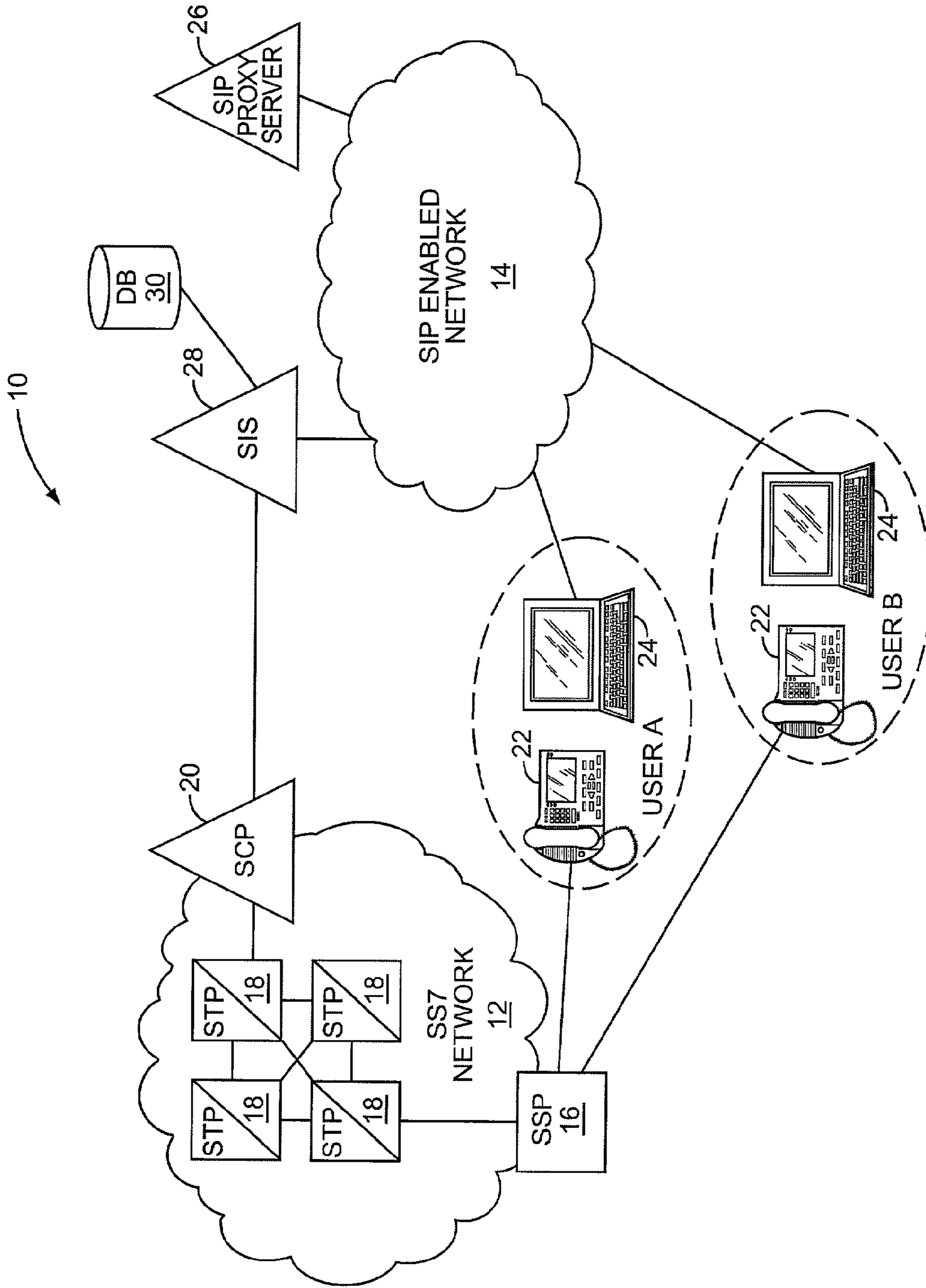


FIG. 1

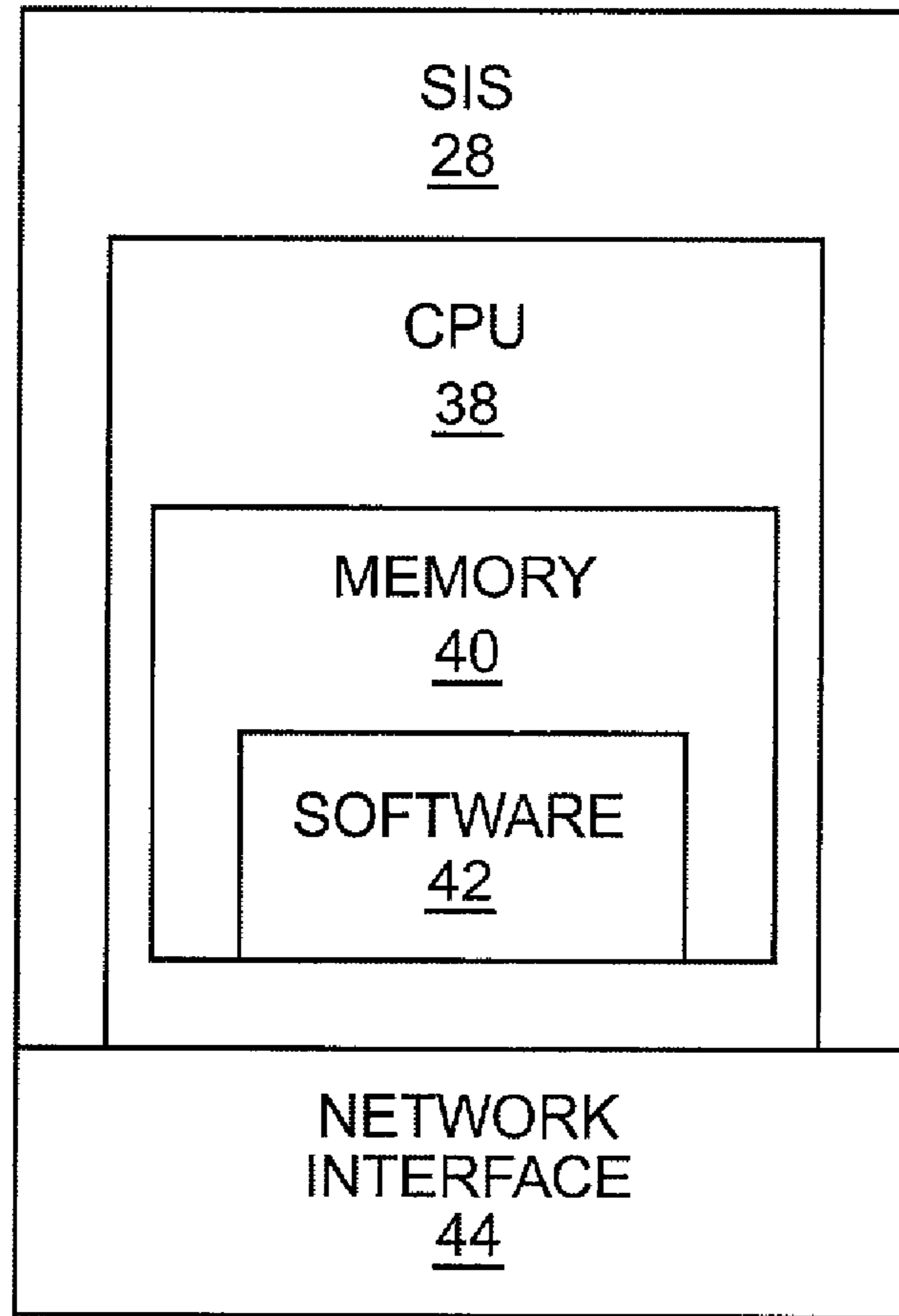


FIG. 2

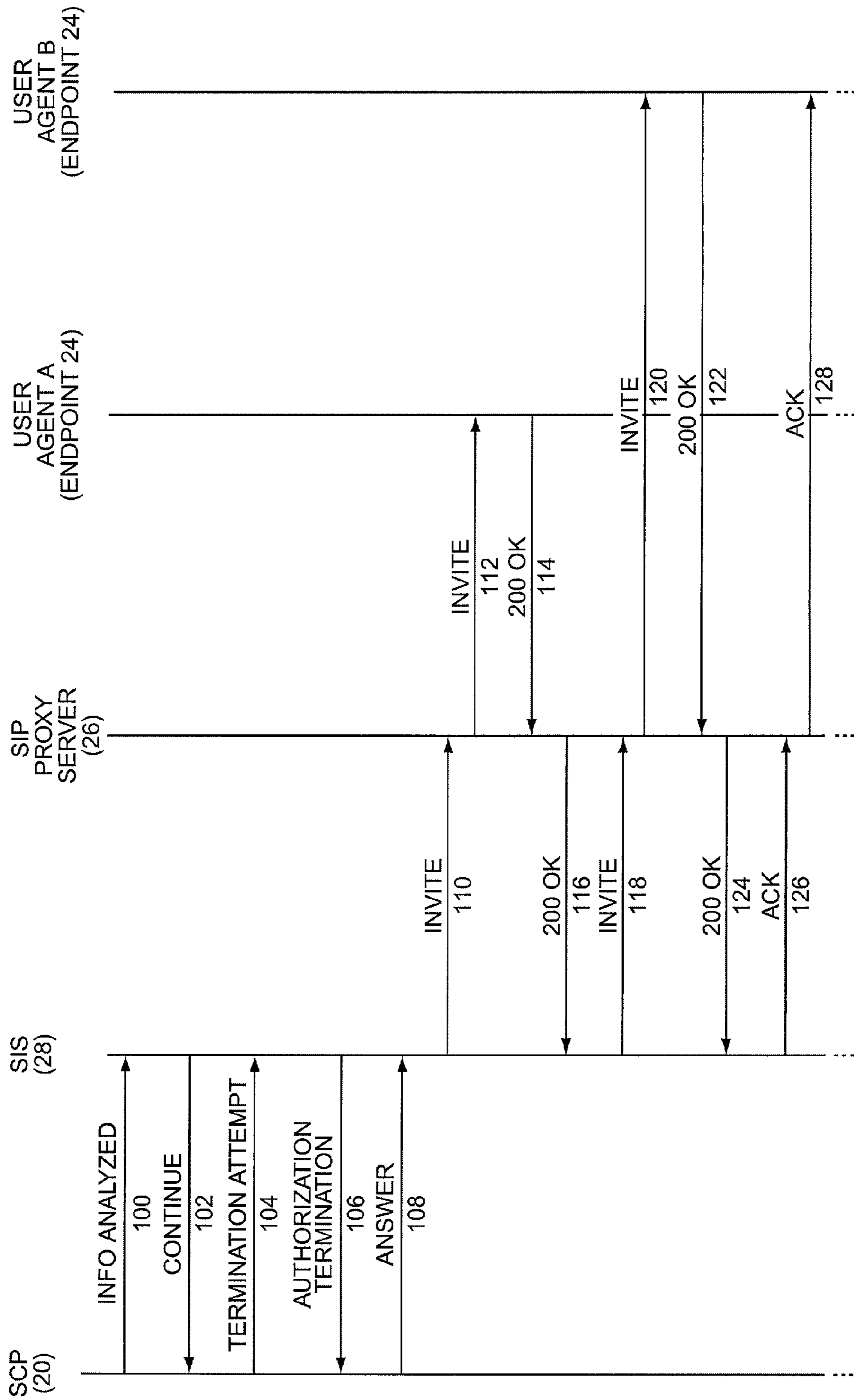


FIG. 3A

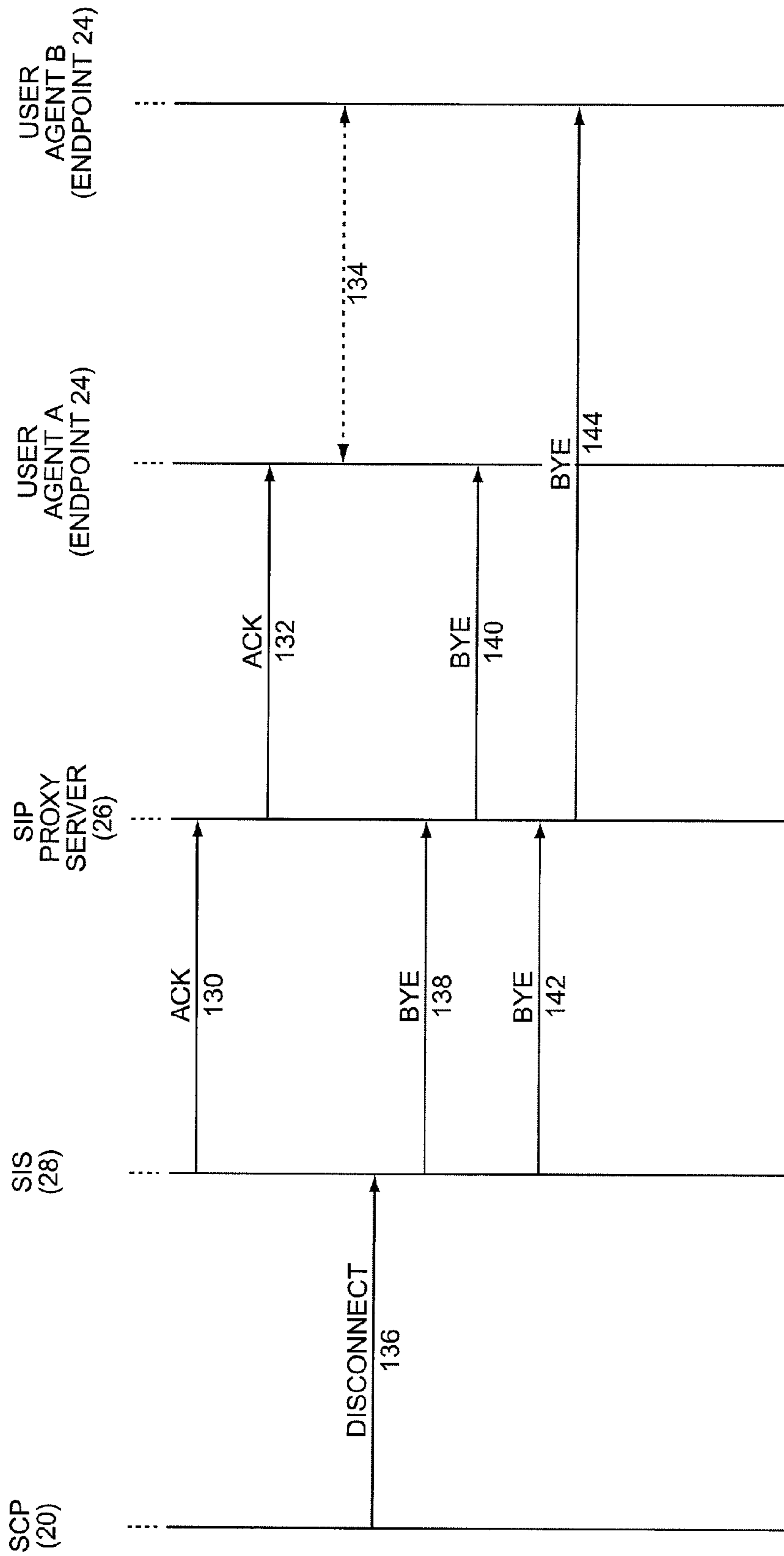


FIG. 3B

1**INTEGRATING MULTIMEDIA
CAPABILITIES WITH CIRCUIT-SWITCHED
CALLS**

This application claims the benefit of Provisional Application Ser. No. 60/308,177, filed Jul. 27, 2001, the disclosure of which is incorporated herein by reference.

FIELD OF THE INVENTION

The present invention relates to multimedia communications, and in particular, relates to integrating multimedia capabilities with circuit-switched calls.

BACKGROUND OF THE INVENTION

The acceptance of network applications and the Internet has given rise for a need to associate voice communications with various network-based applications and functions. While engaged in a telephone conference, users often share applications, such as whiteboarding applications, and web pages to enhance communications. In most instances, users initiate the voice call and then establish an application sharing arrangement independently of the voice call.

Numerous software packages have attempted to integrate voice and data communications over packet-switched networks. Unfortunately, the availability of quality packet-based voice systems is low while the circuit-switched voice systems are widely available. As such, attempts have been made to associate circuit-switched voice calls with packet-switched multimedia applications.

Previous attempts to integrate circuit switched voice calls and multimedia sessions have required proprietary protocols or cumbersome protocols, such as H.323. The lack of flexibility and complexity of these protocols have suppressed their acceptance and availability. Accordingly, there is a need for an efficient and easy to implement technique for integrating circuit-switched voice calls and packet-switched multimedia capability. Further, there is a need to provide such integration without requiring proprietary protocols and by using accepted standards that are readily available.

SUMMARY OF THE INVENTION

The present invention monitors call signaling events stemming from a circuit-switched call between a caller and a called party and controls a packet-session between user agents on respective endpoints associated with the caller and called party. The endpoints may include any type of computational device capable of facilitating the packet-session over a packet-switched network. Control of the user agents may be provided via a proxy for the user agents and may use the session initiation protocol (SIP), or like session control protocol for communications.

Directory numbers for the circuit-switched, customer premise equipment supporting the circuit-switched call may be associated with communication addresses for the endpoints. During operation, the directory numbers for the calling and called party are used to identify the addresses of the respective endpoints. When a circuit-switched call is initiated or established, a call signaling trigger is identified, and the packet session is established between the user agents of the endpoints. Upon completion of the circuit-switched call, a corresponding call signaling trigger is identified, and the packet session is ended.

A dedicated integration service may be used to facilitate interaction with the circuit switched network to identify call-

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signaling events and cooperate with a proxy to control the packet session between the user agents. Alternatively, the integration service may be combined with the proxy alone, or with other network devices.

Those skilled in the art will appreciate the scope of the present invention and realize additional aspects thereof after reading the following detailed description of the preferred embodiments in association with the accompanying drawing figures.

**BRIEF DESCRIPTION OF THE DRAWING
FIGURES**

The accompanying drawing figures incorporated in and forming a part of this specification illustrate several aspects of the invention, and together with the description serve to explain the principles of the invention.

FIG. 1 is an illustration of a communication environment according to one embodiment of the present invention.

FIG. 2 is a block representation of a SIP integration server according to one embodiment of the present invention.

FIGS. 3A and 3B are an exemplary communication flow according one embodiment of the present invention.

**DETAILED DESCRIPTION OF THE PREFERRED
EMBODIMENTS**

The embodiments set forth below represent the necessary information to enable those skilled in the art to practice the invention and illustrate the best mode of practicing the invention. Upon reading the following description in light of the accompanying drawing figures, those skilled in the art will understand the concepts of the invention and will recognize applications of these concepts not particularly addressed herein. It should be understood that these concepts and applications fall within the scope of the disclosure and the accompanying claims.

With reference to FIG. 1, an exemplary communication environment 10 capable of carrying out the concepts of the present invention is illustrated. The communication environment 10 is depicted as including a system signaling 7 (SS7) network and a session initiation protocol (SIP) enabled packet switched network 14. The SIP enabled network 14 may include any type of packet-switched network having devices using SIP to facilitate communications between two or more devices.

The SS7 network 12 is an advanced intelligent network (AIN) capable of providing call signaling for voice-based communications over the public switched telephone network (PSTN) and well known services such as toll-free dialing, automatic redial, call back, and calling number delivery (caller ID). The primary components of a typical SS7 network 12 include service switching points (SSP) 16, service transfer points (STP) 18, and service control points (SCP) 20.

A typical SSP 16, such as Nortel Networks Limited's DMS100, provides standard voice switching and is equipped with SS7 hardware, software, and signaling links. The SSPs 16 are the "end-points" of an SS7 network and typically reside between the customer premise equipment 22 and the SS7 network 12. SSPs 16 also typically provide the end-to-end circuit for the voice transport and perform most of the basic call processing required to setup and terminate voice calls.

The STPs 18 are typically reliable, high-speed packet switches that route SS7 signaling messages throughout the SS7 network 12. The SCPs 20 are computing platforms that run applications providing enhanced service logic to default

call processing actions of the SSPs 16, depending on the application. Those skilled in the art will have an appreciation and understanding of the SS7 network 12 and like signaling networks.

Using AIN concepts, the SSPs 16 will temporarily halt call processing for certain events or points during a call and send information relating to the events to the SCP 20. The SCP 20 will either continue call processing or perform an action, such as release, redirect, or terminate the call. As such, the SCP 20 is routinely engaged in call processing and has information pertaining to the call being processed and events taking place during such processing. The present invention provides a mechanism for the SIP network 14 to interact with the SCP 20 to identify events during call processing as well as influence call processing as desired.

The Internet Engineering Task Force's RFC 2543, which is incorporated in its entirety by reference, provides the ability to establish sessions between endpoints 24 over a packet switched network, such as the SIP network 14, running the internet protocol (IP). Once established, these sessions can exchange media capabilities and set up multiple media paths between the endpoints 24 based on their capabilities. A SIP endpoint 24 will support a User Agent (UA).

User Agents register their ability to receive calls with a SIP proxy server 26 by sending "REGISTER" messages to the SIP proxy server 26. The "REGISTER" message informs the SIP proxy server 26 of the SIP uniform resource locator (URL), which identifies the User Agent to the network. The "REGISTER" message also contains information about how to reach the specific User Agent over the SIP network 14. For instance, the "REGISTER" message may provide the User Agent's IP address and port in which the User Agent will monitor or facilitate communications.

Typically, when a User Agent wants to initiate a call to another User Agent, it will send an "INVITE" message to the SIP proxy server 26 specifying the targeted User Agent in the "TO" header. Identification of a User Agent takes the form of a SIP URL, <username>@<domain>, such as janedoe@nortelnetworks.com. The SIP proxy server 26 will use the SIP URL in the "TO" header of the message to determine if the User Agent is registered. The username is usually unique within the namespace of the specified domain.

If the targeted User Agent has registered, the SIP proxy server 26 will forward the "INVITE" message directly to the targeted User Agent. The User Agent responds with a 200 OK message to the originating User Agent via the SIP proxy server 26, and a session between the two User Agents will be established as per the message exchange required in the SIP specification. Capabilities are passed between the two User Agents as parameters embedded within the session setup messages such as INVITE, 200 OK, and ACK. Media capabilities can also be exchanged using the SIP "INFO" message. Capabilities are typically described using the Session Description Protocol (SDP). Once User Agents are in an active session with each other and understand each other's capabilities, the respective endpoints 24 providing the User Agents can exchange the specified media content.

Reference is made to the Internet Engineering Task Force draft "draft-rosenberg-sip-3 pcc-00.ext" for SIP third party call control, which is outlined below.

Third party call control may take the following form. A central controller first calls one of the participants, A, and presents the INVITE message without any media. When this call is complete, the controller has the SDP needed to communicate with A. The controller then uses the SDP to initiate a call to participant B. When this call is completed, the controller has the SDP needed to communicate with B. This

information is then passed to A. The result is that there is a call leg between the controller and A, a call leg between the controller and B, but media between A and B.

In the preferred embodiment of the invention, an application server is configured to interact with the SS7 network 12 or like circuit switched network via the SCP 20. The application server is generally referred to herein as a "SIP Integration Server" or SIS 28, which monitors call events from the SS7 network 12 and performs third party call control between the Calling and Called User Agents on the SIP network based on those events. The SIS 28 may be affiliated with a database 30 to provide information for operation and pertaining to the directory numbers, addresses and the like for User Agents and CPEs 22.

As shown in FIG. 2, the SIS 28 may be a typical web server having a central processing unit (CPU) 38 with the requisite memory 40 containing the software 42 and data necessary for operation. The CPU 38 is associated with a network interface 44 facilitating communications with other devices, such as the endpoints 24, SIP proxy server 26, SCP 20, and the database 30 through any number of local area networks, routers, switches and hubs in traditional fashion.

Subscribers to the SIS 28 will have certain AIN triggers provisioned against their directory number for the CPE 22 within their local SSP 16. The triggers correspond to events during call processing. Whenever the subscriber is involved with a telephone call, the SSP 16 will route the appropriate triggers to the SIS service for interpretation. It should be noted that this disclosure is not tied to the mechanism by which AIN messages are routed to the SIS 28. In addition, SIS subscribers will have User Agents installed on their personal computer or other computing device that has access to the SIP network 14. These User Agents can be any SIP enabled applications that know how to register with the SIP proxy server 26, can establish SIP sessions, and can exchange multi-media content of one or more types. For another example of a SIP user agent benefiting from the present invention, please refer to U.S. patent application Ser. No. 09/666,583, filed Sep. 21, 2001, entitled "AUTOMATED WEB BROWSER SYNCHRONIZATION," the disclosure of which is incorporated herein by reference.

SIP User Agents register with the SIP proxy server 26 as specified in the SIP specification and preferably identify their username in the SIP URL to be equivalent to the directory number that is assigned to their telephone. For example if John Smith has a directory number of (555) 991-1234, then John's SIP User Agent would register with the username of <5559911234>@<domain> with the SIP proxy server 26. Such registration is preferred because the directory numbers are the main identifier used within the AIN triggers received by the SIS 28 from the SSP 16.

When the SIS 28 needs to interact with the SIP network 14, it will send SIP messages to the SIP proxy server 26 and identify the intended recipient of the message using the corresponding directory number. The SIP proxy server 26 will forward the message to the appropriate User Agent. Notably, this disclosure depicts the SIP Integration Server and the SIP Proxy as two separate entities; however, the functionality of both may reside within the same platform or even within the same application. Thus, the concepts of the present invention may be implemented with the proxy functionality embedded within the SIS 28, and vice versa.

A SIP session is initiated when the SIS 28 receives a trigger from the SCP 20 that the voice call for its user has been answered. On receiving this trigger from the SCP 20, the SIS 28 first attempts to establish a SIP session with the caller's User Agent without specifying any media capabilities. The

caller's User Agent accepts the session invitation and will respond with its capability descriptions. The SIS 28 then attempts to establish a session with the called user's User Agent and sends the capabilities of the caller's User Agent. The called User Agent accepts the session invitation and responds with its own capability descriptions. The SIS 28 then forwards the called User Agents capability descriptions back to the caller's User Agent to complete the capability negotiations, and the two User Agents begin exchanging content on their established media path(s).

The SIP Session is terminated whenever the SIS 28 receives a Disconnect trigger from the SCP 20 indicating that either side has released the voice call. Once this notification is received, the SIS 28 sends a SIP BYE message to both User Agents and the SIP Session is terminated.

Turning now to FIGS. 3A and 3B, an exemplary communication flow is described for setting up and releasing calls using the SIS 28 to synchronize SIP multi-media sessions with voice calls. Those skilled in the art will recognize the exemplary communication flow highlights only some of the applications made possible by the present invention. The concepts and architecture of the present invention allow for numerous service extensions, which should become apparent to those skilled in the art upon reading this disclosure.

For the exemplary flow of FIGS. 3A and 3B, User A places a call to User B and a SIP Session is established with multi-media capability, such as whiteboarding or application sharing. Assume User A has a telephone number of (555) 444-1111 and a SIP User Agent A application running on an associated personal computer (endpoint 24) that supports whiteboarding. Further assume that User Agent A has previously registered with the SIP proxy server 26 as 5554441111@<service_provider_domain>. User B has a telephone number of (555) 333-2222 and a SIP User Agent B application running on his/her personal computer that also supports whiteboarding. The User Agent for User B has previously registered with the SIP proxy server 26 as 5553332222@<service_provider_domain>.

Initially, User A uses his CPE 22 (i.e., telephone) to place a call to User B by dialing 333-2222. User A's SSP 16 sends an InfoAnalyzed AIN trigger to the SCP 20 servicing the SIS 28. The InfoAnalyzed trigger contains both the directory number of the calling user (User A) and the directory number of the called user (User B). The SCP 20 forwards the InfoAnalyzed trigger to the SIS 28 (step 100). The SIS 28 looks up the subscriber profile for User A on database 30 by indexing on the calling user's (User A's) directory number. The SIS 28 finds the profile for User A and maintains the knowledge that User A is placing a call to User B.

The SIS 28 replies to the SCP with a Continue response thus allowing the SSP 16 to continue to setup the call to User B's CPE 22 (step 102). The SSP 16 servicing User B's CPE 22 sends a TerminationAttempt trigger to the SCP 20, which forwards it on to the SIS 28 (step 104). The SIS 28 looks up the subscriber profile for User B by indexing on the called user's (User B) directory number. The SIS 28 finds the profile for User B and maintains the knowledge that User A is placing a call to User B.

The SIS 28 replies by sending an AuthorizeTermination response back to the SSP 16 via the SCP 20 (step 106). The SSP 16 terminates the call to User B's CPE 22, which begins to ring. User B answers the call by lifting the handset of the respective CPE 22. The SSP 16 servicing User B's CPE 22 sends an Answer trigger to the SIS 28 via the SCP 20 (step 108). The SIS 28 now knows that the voice call between User A and User B has been answered.

To set up an associated SIP session, the SIS 28 establishes a session with the caller's (User A) User Agent first. To do this, the SIS 28 sends an "INVITE" message to the SIP proxy server 26 with the username in the TO: field set to the telephone number of User A—(5554441111@<service_provider_domain>) (step 110). This initial INVITE message typically does not contain any capability information. The SIP proxy server 26 forwards the message on to the User Agent on endpoint 24 for User A (step 112).

User A's User Agent replies with a 200 OK message and specifies its capability information, such as media type and coding/decoding (CODEC) support, in the message body (step 114). The SIP proxy server 26 forwards the 200 OK back to the SIS 28 (step 116).

The SIS 28 sends an "INVITE" message to the SIP proxy server 26 with the username of the TO: field set to the telephone number of User B (5553332222@<service_provider_domain>) (step 118). Also included in this message is the capability description received in the 200 OK message from the caller's (User A) User Agent. The SIP proxy server 26 forwards the message on to the User Agent on endpoint 24 for User B (step 120).

The User Agent for User B replies with a 200 OK message and specifies its capability information in the message body (step 122). The SIP proxy server 26 forwards the 200 OK response back to the SIS 28 (step 124). The SIS 28 acknowledges the 200 OK response from the User Agent of User B by sending an ACK message to User Agent for User B via the SIP proxy server 26 (steps 126 and 128).

The SIS 28 now needs to send the capability information it received from User B's User Agent to User A's User Agent. Accordingly, the SIS 28 builds an ACK message with the capability description received from the 200 OK message from User Agent of User B and sends the ACK message to User A's User Agent via the SIP proxy server 26 (steps 130 and 132). Media capability information could also have been sent in a second "INVITE" message as opposed to the ACK as described in association with the Third Party Call Control IETF draft discussed above.

A SIP session is now set up between the User Agents for User A and User B, thus allowing users to perform the multimedia function(s) as described within the session description of the message body. When either user ends the voice call by hanging up their handset of the CPE 22, the SCP 20 will send an appropriate Disconnect trigger to the SIS 28 (step 136). The SIS 28 will terminate the SIP Session by sending "BYE" messages to each User Agent via the SIP proxy server 26 (steps 138 through 144).

Those skilled in the art will recognize improvements and modifications to the preferred embodiments of the present invention. All such improvements and modifications are considered within the scope of the concepts disclosed herein and the claims that follow.

What is claimed is:

1. A method for integrating separate circuit-switched voice and packet-switched data sessions comprising:
 - a. identifying call signaling events for a separate circuit-switched call between a caller and a called party over a circuit-switched network; and
 - b. controlling a separate packet-switched session for a packet-session between a caller user agent for a caller endpoint and a called party user agent on a called party endpoint based on at least one of the call signaling events from the separate circuit-switched call, wherein the caller and called party are associated with the caller and called party endpoints, such that the caller and called party may conduct concurrent voice and data sessions.

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2. The method of claim 1 wherein the controlling step comprises communicating with the caller and called party user agents via a proxy for the caller and called party user agents.

3. The method of claim 1 further comprising registering the caller and called party with a proxy using directory numbers associated with the caller and called party.

4. The method of claim 1 wherein the controlling step is effected using session initiation protocol (SIP).

5. The method of claim 1 wherein the packet-session is a SIP session.

6. The method of claim 1 wherein the identifying step comprises monitoring triggers corresponding to the call signaling events provided by a call signaling control system in the circuit-switched network.

7. The method of claim 1 further comprising:

- a. identifying directory numbers for the caller and called party; and
- b. determining addresses of the caller and called party user agents to use for the packet-session based on the directory numbers.

8. The method of claim 7 wherein the addresses for the caller and called party user agents include uniform resource locators corresponding to the directory numbers for the caller and called party.

9. The method of claim 1 wherein:

- a. the identifying step comprises identifying a call signaling trigger representing initiation or establishment of the circuit-switched call between the caller and the called party; and
- b. the controlling step comprises initiating the packet-session between the caller user agent and the called party user agent upon identifying the call signaling trigger representing the initiation or establishment of the circuit-switched call between the caller and the called party.

10. The method of claim 1 wherein:

- a. the identifying step further comprises identifying a call signaling trigger representing completion of the circuit-switched call between the caller and the called party; and
- b. the controlling step comprises ending the packet-session between the caller user agent and the called party user agent upon identifying the call signaling trigger representing completion of the circuit-switched call between the caller and the called party.

11. The method of claim 1 wherein communications with the caller and called party user agents are facilitated using session initiation protocol (SIP) via a SIP proxy for the caller and called party user agents and:

- a. the identifying step comprises:
 - i. identifying a call signaling trigger representing initiation or establishment of the circuit-switched call between the caller and the called party; and
 - ii. identifying a call signaling trigger representing completion of the circuit-switched call between the caller and the called party; and
- b. the controlling step comprises:
 - i. initiating the packet-session between the caller user agent and the called party user agent upon identifying the call signaling trigger representing initiation or establishment of the circuit-switched call between the caller and the called party; and
 - ii. ending the packet-session between the caller user agent and the called party user agent upon identifying the call signaling trigger representing completion of the circuit-switched call between the caller and the called party.

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12. A system for integrating circuit-switched voice and packet-switched sessions comprising:

- a. a communication interface; and
- b. a control system associated with the communication interface, the control system configured to:
 - i. identify call signaling events for a circuit-switched call between a caller and a called party over a circuit-switched network; and
 - ii. control a packet-switched session for a packet-session between a caller user agent for a caller endpoint and a called party user agent on a called party endpoint based on at least one of the call signaling events, wherein the caller and called party are associated with the caller and called party endpoints.

13. The system of claim 12 wherein the control system is further configured to communicate with the caller and called party user agents via a proxy for the caller and called party user agents.

14. The system of claim 12 wherein the control system is further configured to effect communications using session initiation protocol (SIP).

15. The system of claim 12 wherein the packet session is a SIP session.

16. The system of claim 12 wherein the control system is further configured to monitor triggers corresponding to the call signaling events provided by a call signaling control system in the circuit-switched network.

17. The system of claim 12 wherein the control system is further configured to:

- a. identify directory numbers for the caller and called party; and
- b. determine addresses of the caller and called party user agents to use for the packet-session based on the directory numbers.

18. The system of claim 17 wherein the addresses for the caller and called party user agents include uniform resource locators corresponding to the directory numbers for the caller and called party.

19. The system of claim 12 wherein the control system is further configured to:

- a. identify the call signaling events by identifying a call signaling trigger representing initiation or establishment of the circuit-switched call between the caller and the called party, and
- b. control the packet-switched session by initiating the packet-session between the caller user agent and the called party user agent upon identifying the call signaling trigger representing the initiation or establishment of the circuit-switched call between the caller and the called party.

20. The system of claim 12 wherein the control system is further configured to:

- a. identify the call signaling events by identifying a call signaling trigger representing completion of the circuit-switched call between the caller and the called party; and
- b. control the packet-switched session by ending the packet-session between the caller user agent and the called party user agent upon identifying the call signaling trigger, representing the completion of the circuit-switched call between the caller and the called party.

21. The system of claim 12 wherein communications with the caller and called party user agents are facilitated using session initiation protocol (SIP) via a SIP proxy for the caller and called party user agents and the control system is further configured to:

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- a. identify the call signaling events by:
- i. identifying a call signaling trigger representing initiation or establishment of the circuit-switched call between the caller and the called party; and
 - ii. identifying a call signaling trigger representing completion of the circuit-switched call between the caller and the called party; and
- b. control the packet-switched session by:
- i. initiating the packet-session between the caller user agent and the called party user agent upon identifying the call signaling trigger representing the initiation or establishment of the circuit-switched call between the caller and the called party; and
 - ii. ending the packet-session between the caller user agent and the called party user agent upon identifying the call signaling trigger, representing the completion of the circuit-switched call between the caller and the called party.
- 22.** A computer readable medium with software for integrating circuit-switched voice and packet-switched sessions, the software embedded in the computer readable medium and comprising instructions for a computer to:
- a. identify call signaling events for a circuit-switched call between a caller and a called party over a circuit-switched network; and
 - b. control a packet-switched session for a packet-session between a caller user agent for a caller endpoint and a called party user agent on a called party endpoint based on at least one of the call signaling events, wherein the caller and called party are associated with the caller and called party endpoints.

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- 23.** The computer readable medium of claim **22** further comprising instructions to:
- a. identify the call signaling events by identifying a call signaling trigger representing initiation or establishment of the circuit-switched call between the caller and the called party; and
 - b. control the packet-switched session by initiating the packet-session between the caller user agent and the called party user agent upon identifying the call signaling trigger representing the initiation or establishment of the circuit-switched call between the caller and the called party.
- 24.** The computer readable medium of claim **22** further comprising instructions to:
- a. identify the call signaling events by identifying a call signaling trigger, representing completion of the circuit-switched call between the caller and the called party; and
 - b. control the packet-switched session by ending the packet-session between the caller user agent and the called party user agent upon identifying the call signaling trigger representing the completion of the circuit-switched call between the caller and the called party.
- 25.** The computer readable medium of claim **22** further comprising instructions to communicate with the caller and called party user agents via a proxy for the caller and called party user agents.
- 26.** The computer readable medium of claim **22** further comprising instructions to effect communications using session initiation protocol (SIP).
- 27.** The computer readable medium of claim **26** wherein the packet-session is a SIP session.

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